

AMENDMENTS TO THE CLAIMS

This listing of claims will replace all prior versions, and listings, of claims in the application:

Listing of Claims:

- 1 1. (Currently amended) An apparatus that provides a unified
2 telephony solution, comprising:
3 an application server configured to provide telephony services;
4 a voice extensible markup language (VXML) browser configured to access
5 telephony services through the application server;
6 a telephony controller configured to access telephony services through the
7 VXML browser, wherein the telephony controller includes a SIP framework with
8 a SIP servlet container, wherein the SIP servlet container includes a plurality of
9 SIP servlets for interfacing with a SIP network; and
10 a telephony gateway that provides an interface to a public switched
11 telephone network (PSTN).
- 1 2. (Canceled).
- 1 3. (Currently amended) The apparatus of ~~claim 2~~ claim 1, wherein a
2 new telephony service can be added by including a new SIP servlet in the SIP
3 servlet container.

1 4. (Currently amended) The apparatus of ~~claim 2~~ claim 1, wherein the
2 plurality of SIP servlets are registered with a remote method invocation (RMI)
3 registry.

1 5. (Currently amended) The apparatus of ~~claim 2~~ claim 1, wherein the
2 telephony services provide at least two of:
3 a telephone system;
4 a call center;
5 an interactive voice response (IVR) system; and
6 a voicemail system.

1 6. (Currently amended) The apparatus of ~~claim 2~~ claim 1, wherein the
2 apparatus operates using a Voice Over Internet Protocol (VOIP).

1 7. (Currently amended) The apparatus of ~~claim 2~~ claim 1, wherein the
2 application server is coupled to a database that provides access to the plurality of
3 SIP servlets.

1 8. (Currently amended) A method that provides a unified telephony
2 solution, comprising:
3 receiving a request for a telephony service at a telephony controller,
4 wherein the telephony controller includes a SIP framework with a SIP servlet
5 container, wherein the SIP servlet container includes a plurality of SIP servlets for
6 interfacing with a SIP network; and
7 in response to the request, accessing a telephony service provided by an
8 application server;

9 wherein the application server is accessed through a voice extensible
10 markup language (VXML) browser;
11 wherein performing the telephony service involves interfacing to a public
12 switched telephone network (PSTN) through a telephony gateway.

1 9. (Canceled)

1 10. (Currently amended) The method of ~~claim 9~~ claim 8, wherein a
2 new telephony service can be added by including a new SIP servlet in the SIP
3 servlet container.

1 11. (Currently amended) The method of ~~claim 9~~ claim 8, wherein the
2 plurality of SIP servlets are registered with a remote method invocation (RMI)
3 registry.

1 12. (Currently amended) The method of ~~claim 9~~ claim 8, wherein the
2 telephony services provide at least two of:
3 a telephone system;
4 a call center;
5 an interactive voice response (IVR) system; and
6 a voicemail system.

1 13. (Currently amended) The method of ~~claim 9~~ claim 8, wherein the
2 telephony services operate using the Voice Over Internet Protocol (VOIP).

1 14. (Currently amended) The method of ~~claim 9~~ claim 8, wherein the
2 application server is coupled to a database that provides access to the plurality of
3 SIP servlets.

1 15. (Currently amended) A computer-readable storage ~~medium~~ device
2 storing instructions that when executed by a computer cause the computer to
3 perform a method that provides a unified telephony solution, the method
4 comprising:
5 receiving a request for a telephony service at a telephony controller,
6 wherein the telephony controller includes a SIP framework with a SIP servlet
7 container, wherein the SIP servlet container includes a plurality of SIP servlets for
8 interfacing with a SIP network; and
9 in response to the request, accessing a telephony service provided by an
10 application server through a voice extensible markup language (VXML) browser;
11 wherein the telephony service involves interfacing to a public switched
12 telephone network (PSTN) through a telephony gateway.

1 16. (Canceled)

1 17. (Currently amended) The computer-readable storage
2 ~~medium~~ device of ~~claim 16~~ claim 15, wherein a new telephony service can be
3 added by including a new SIP servlet in the SIP servlet container..

1 18. (Currently amended) The computer-readable storage
2 ~~medium~~ device of ~~claim 16~~ claim 15, wherein the plurality of SIP servlets are
3 registered with a remote method invocation (RMI) registry.

1 19. (Currently amended) The computer-readable storage
2 ~~medium~~device of ~~claim 16~~ claim 15, wherein the telephony services provide at
3 least two of:
4 a telephone system;
5 a call center;
6 an interactive voice response (IVR) system; and
7 a voicemail system.

1 20. (Currently amended) The computer-readable storage
2 ~~medium~~device of ~~claim 16~~ claim 15, wherein the telephony services operate using
3 the Voice Over Internet Protocol (VOIP).

1 21. (Currently amended) The computer-readable storage
2 ~~medium~~device of ~~claim 16~~ claim 15, wherein the application server is coupled to a
3 database that provides access to the plurality of SIP servlets.